AMENDMENTS TO THE SPECIFICATION

Please replace the paragraph beginning on page 7 line 20 with the following:

Figure 1 is a block diagram showing an audio delivery system 100 that overcomes the limitations of the prior art and provides a flexible method for streaming an encoded multi-channel audio format over the Internet. In Figure 1, one or more audio sources 101 are provided, typically through a communication network 102, to a computer 103 operated by a listener 148. The computer 103 receives the audio data, decodes the data if necessary, and provides the audio data to one or more loudspeakers, such as, loudspeakers 146, ~~148~~147, or to a multi-channel loudspeaker system (not shown). The audio sources 101 can include, for example, a Circle Surround 5.1 encoded source 110, a Dolby Surround encoded source 111, a conventional two-channel stereo source 112 (encoded as raw audio, MP3 audio, RealAudio, WMA audio, etc.), and/or a single-channel monaural source 113. In one embodiment, the computer 103 includes a decoder 104 for Circle Surround 5.1, and, optionally, an enhanced signal processing module 105 (e.g., an SRS Laboratories TruSurround system and/or an SRS Laboratories WOW system as described in connection with Figures 4-17). The signal processing module 105 is useful for a wide variety of systems. In particular, the signal processing module 105 incorporating TruSurround and/or WOW is particularly useful when the computer 103 is connected to the two-channel speaker system 146, 147. The signal processing module 105 incorporating TruSurround and/or WOW is also particularly useful when the speakers 146 and 147 are not optimally placed or do not provide optimal bass response.

Please replace the paragraph beginning on page 19 line 9 with the following:

Figure 10 is block diagram of the stereo image correction system 422, which inputs the left and right stereo signals 426 and 428. The image-correction system 422 corrects the distorted spectral densities of various sound systems by advantageously dividing the audible frequency spectrum into a first frequency component, containing relatively lower frequencies, and a second frequency component, containing relatively higher frequencies. Each of the left and right signals 426 and 428 is separately processed through corresponding low-frequency correction systems 1080, 1082, and high-frequency correction systems 1084 and 1086. It should be pointed out that in one embodiment the

correction systems 1080 and 1082 will operate in a relatively "low" frequency range of approximately 100 Hz to 1000 ~~Hertz~~Hz, while the correction systems 1084 and 1086 will operate in a relatively "high" frequency range of approximately 1000 Hz to 10,000 ~~Hertz~~Hz. This is not to be confused with the general audio terminology wherein low frequencies represent frequencies up to 100 ~~Hertz~~Hz, mid frequencies represent frequencies between 100 Hz to 4 kHz, and high frequencies represent frequencies above 4 kHz.

Please replace the paragraph beginning on page 20 line 5 with the following:

Once the sound source is properly positioned through energy correction of the audio signal, the bass enhancement unit 101 corrects for low frequency deficiencies in the loudspeakers 146, 147 and provides bass-corrected left and right channel signals to the stereo enhancement system 424. The stereo enhancement system 424 conditions the stereo signals to broaden (horizontally) the stereo image emanating from the apparent sound source. As will be discussed in conjunction with Figures 8A and 8B, the stereo image enhancement system 424 can be adjusted through a stereo orientation device to compensate for the actual location of the sound source.

Please replace the paragraph beginning on page 20 line 15 with the following:

The left and right signals 1094, 1096 provided from the bass enhancement unit 401 are inputted by the enhancement system 424 and provided to a difference-signal generator 1001 and a sum signal generator 1004. A difference signal (L_c-R_c) representing the stereo content of the corrected left and right input signals, is presented at an output 1002 of the difference signal generator 1001. A sum signal, (L_c+R_c) representing the sum of the corrected left and right stereo signals is generated at an output 1006 of the sum signal generator 1004.

Please replace the paragraph beginning on page 21 line 8 with the following:

The shaped difference signal 1040 is provided to a mixer 1042, which also receives the sum signal from the device ~~1006~~1008. In one embodiment, the stereo signals 1094 and 1096 are also provided to the mixer 1042. All of these signals are

combined within the mixer 1042 to produce an enhanced and spatially-corrected left output signal 1030 and right output signal 1032.

Please replace the paragraph beginning on page 21 line 26 with the following:

To those skilled in the art, a typical filter is usually characterized by a pass-band and stop-band of frequencies separated by a cutoff frequency. The correction curves, of Figures 6A-6C, although representative of typical signal filters, can be characterized by a pass-band, a stop-band, and a transition band. A filter constructed in accordance with the characteristics of Figure 6A has a pass-band above approximately 1000 Hz, a transition-band between approximately 100 and 1000 Hz, and a stop-band below approximately 100 Hz. Filters according to ~~figures~~ Figure 6B ~~and 6C~~ have pass-bands above approximately 10 kHz, transition-bands between approximately 1 kHz and 10 kHz, and a stop-band below approximately 1 kHz. Filters according to Figure 6C have a stop-band above approximately 10 kHz, transition-bands between approximately 1 kHz and 10 kHz, and pass-bands below approximately 1 kHz. In one embodiment, the filters are first-order filters.

Please replace the paragraph beginning on page 22 line 4 with the following:

As can be seen in Figures 6A-6C, spatial correction of an audio signal by the systems 1080, 1082, 1084, and 1086 is substantially uniform within the pass-bands, but is largely frequency-dependent within the transition bands. The amount of acoustic correction applied to an audio signal can be varied as a function of frequency through adjustment of the stereo image correction system, which varies the slope of the transition bands of Figures 6A-6C. As a result, frequency-dependent correction is applied to a first frequency range between 100 Hz and 1000 ~~hertz~~Hz, and applied to a second frequency range of 1000 Hz to 10,000 ~~hertz~~Hz. An infinite number of correction curves are possible through independent adjustment of the correction systems 1080, 1082, 1084 and 1086.

Please replace the paragraph beginning on page 22 line 13 with the following:

In accordance with one embodiment, spatial correction of the higher frequency stereo-signal components occurs between approximately 1000 Hz and 10,000 Hz. Energy correction of these signal components may be positive, i.e., boosted, as depicted

in Figure 6B, or negative, i.e., attenuated, as depicted in Figure 6C. The range of boost provided by the correction systems 1084, 1086 is characterized by a maximum-boost curve 660 and a minimum-boost curve ~~112~~662. Curves 664, 666, and 668 represent still other levels of boost, which may be required to spatially correct sound emanating from different sound reproduction systems. Figure 6C depicts energy-correction curves that are essentially the inverse of those in Figure 6B.

Please replace the paragraph beginning on page 22 line 22 with the following:

Since the lower frequency and higher frequency correction factors, represented by the curves of Figures 6A-6C, are added together, there is a wide range of possible spatial correction curves applicable between the frequencies of 100 to 10,000 Hz. Figure 6D is a graphical representation depicting a range of composite spatial correction characteristics provided by the stereo image correction system ~~1022~~422. Specifically, the solid line curve 680 represents a maximum level of spatial correction comprised of the curve 650 (shown in Fig. 6A) and the curve 660 (shown in Fig. 6B). Correction of the lower frequencies may vary from the solid curve 680 through the range designated by θ_1. Similarly, correction of the higher frequencies may vary from the solid curve 680 through the range designated by θ_2. Accordingly, the amount of boost applied to the first frequency range of 100 Hz to 1000 ~~Hertz~~ Hz varies between approximately 0 and 15 dB, while the correction applied to the second frequency range of 1000 to 10,000 Hertz may vary from approximately ~~13~~ 15 dB to ~~-15~~30 dB.

Please replace the paragraph beginning on page 23 line 24 with the following:

According to one embodiment, the range for the perspective curves of Figure 7 is defined by a maximum gain of approximately 10-15 dB located at approximately 125 to 150 Hz. The maximum gain values denote a turning point for the curves of Figure 7 whereby the slopes of the curves 790, 792, 794, 796, and 798 change from a positive value to a negative value. Such turning points are labeled as points A, B, C, D, and E in Figure 7. The gain of the perspective curves decreases below 125 Hz at a rate of approximately 6 dB per octave. Above 125 Hz, the gain of the curves of Figure 7 also decreases, but at variable rates, towards a minimum-gain turning point of approximately -2 to +10 dB. The minimum-gain turning points vary significantly between the curves 790,

792, 794, 796, and 798. The minimum-gain turning points are labeled as points A', B', C', D', and E', respectively. The frequencies at which the minimum-gain turning points occur varies from approximately 2.1 kHz for curve 790 to approximately ~~10~~ 5 kHz for curve 798. The gain of the curves 790, 792, 794, 796, and 798 increases above their respective minimum-gain frequencies up to approximately 10 ~~KHz~~kHz. Above 10 ~~KHz~~kHz, the gain applied by the perspective curves begins to level off. An increase in gain will continue to be applied by all of the curves, however, up to approximately ~~120~~ 20 ~~KHz~~kHz, i.e., approximately the highest frequency audible to the human ear.

Please replace the paragraph beginning on page 24 line 17 with the following:

Equalization of the difference signal in accordance with the curves of Figure 7 is intended to boost the difference signal components of statistically lower intensity without overemphasizing the higher-intensity difference signal components. The higher-intensity difference signal components of a typical stereo signal are found in a mid-range of frequencies between approximately 1 kHz to 4 kHz. The human ear has a heightened sensitivity to this same mid-range of frequencies. Accordingly, the enhanced left and right output signals 1030 and 1032 produce a much improved audio effect because ambient sounds are selectively emphasized to fully encompass a listener within a reproduced sound stage.

Please replace the paragraph beginning on page 25 line 12 with the following:

Figure 8A depicts a top view of a sound reproduction environment having loudspeakers 800 and 802 placed slightly forward of, and pointed towards, the sides of a listener 804. The loudspeakers 800 and 802 are also placed below the listener 804 at a elevational position similar to that of the loudspeakers 146, 147 shown in Figure 2. Reference planes A and B are aligned with ears 806, 808 of the listener 804. The planes A and B are parallel to the listener's line-of-sight as shown.

Please replace the paragraph beginning on page 26 line 9 with the following:

A typical small loudspeaker system used for multimedia computers, automobiles, small stereophonic systems, portable stereophonic systems, headphones, and the like, will have an acoustic output response that rolls off at about

150 Hz. Figure 9 shows a curve 906 corresponding approximately to the frequency response of the human ear. Figure 9 also shows the measured response 908 of a typical small computer loudspeaker system that uses a high-frequency driver (tweeter) to reproduce the high frequencies, and a four-inch midrange-bass driver (woofer) to reproduce the midrange and bass frequencies. Such a system employing two drivers is often called a two-way system. Loudspeaker systems employing more than two drivers are known in the art and will work with the present invention. Loudspeaker systems with a single driver are also known and will work with the present invention. The response 908 is plotted on a rectangular plot with an X-axis showing frequencies from ~~15~~ 20 Hz to ~~15~~ 20 kHz. This frequency band corresponds to the range of normal human hearing. The Y-axis in Figure 9 shows normalized amplitude response from 0 dB to -50 dB. The curve 908 is relatively flat in a midrange frequency band from approximately 2 kHz to 10 kHz, showing some roll off above 10 kHz. In the low frequency ranges, the curve 908 exhibits a low-frequency roll off that begins in a midbass band between approximately 150 Hz and 2 kHz such that below 150 Hz, the loudspeaker system produces very little acoustic output.

Please replace the paragraph beginning on page 28 line 15 with the following:

In response to an increase in the amplitude of the envelope of the signal provided to the input of the bass punch unit 1120, the servo loop increases the forward gain of the bass punch unit 1120. Conversely, in response to a decrease in the amplitude of the envelope of the signal provided to the input of the bass punch unit 1120, the servo loop ~~increases~~ decreases the forward gain of the bass punch unit 1120. In one embodiment, the gain of the bass punch unit 1120 increases more rapidly that the gain decreases. Figure 11 is a time domain plot that illustrates the gain of the bass punch unit 1120 in response to a unit step input. One skilled in the art will recognize that Figure 11 is a plot of gain as a function of time, rather than an output signal as a function of time. Most amplifiers have a gain that is fixed, so gain is rarely plotted. However, the Automatic Gain Control (AGC) in the bass punch unit 1120 varies the gain of the bass punch unit 1120 in response to the envelope of the input signal.

Please replace the paragraph beginning on page 29 line 3 with the following:

The attack time constant 1104 and the decay time constant 1106 are desirably selected to provide enhancement of the bass frequencies without overdriving other components of the system such as the amplifier and loudspeakers. Figure 12 is a time-domain plot 1200 of a typical bass note played by a musical instrument such as a bass guitar, bass drum, synthesizer, etc. The plot 1200 shows a higher-frequency portion ~~1240~~ 1244 that is amplitude modulated by a lower-frequency portion having a modulation envelope 1242. The envelope 1242 has an attack portion 1246, followed by a decay portion 1247, followed by a sustain portion 1248, and finally, followed by a release portion 1249. The largest amplitude of the plot 1200 is at a peak 1250, which occurs at the point in time between the attack portion 1246 and the decay portion 1247.

Please replace the paragraph beginning on page 29 line 27 with the following:

Similarly, a drumhead, when struck, will produce an initial set of large excursion vibrations corresponding to the attack portion 1246 and the decay portion 1247. After the large excursion vibrations have died down (corresponding to the end of the decay portion ~~1217~~1247) the drumhead will continue to vibrate for a period of time corresponding to the sustain portion 1248 and release portion 1249. Many musical instrument sounds can be created merely by controlling the length of the periods 1246-1249.

Please replace the paragraph beginning on page 30 line 4 with the following:

As described in connection with Figure 12, the amplitude of the higher-frequency signal is modulated by a lower-frequency tone (the envelope), and thus, the amplitude of the higher-frequency signal varies according to the frequency of the lower frequency tone. The non-linearity of the ear will partially demodulate the signal such that the ear will detect the low-frequency envelope of the higher-frequency signal, and thus produce the perception of the low-frequency tone, even though no actual acoustic energy was produced at the lower frequency. The detector effect can be

enhanced by proper signal processing of the signals in the midbass frequency range, typically between 100 Hz -150 Hz on the low end of the range and 150 Hz -500 Hz on the high end of the range. By using the proper signal processing, it is possible to design a sound enhancement system that produces the perception of low-frequency acoustic energy, even when using loudspeakers that are incapable of producing such energy.

Please replace the paragraph beginning on page 31 line 14 with the following:

Figure 13 is a signal processing block diagram of the bass enhancement system 401 that provides bass enhancement using a peak compressor to control the amplitude of pulses, such as the initial pulse, bass notes. In the system 401, a peak compressor 1302 is interposed between the combiner ~~1418~~ 1318 and the punch unit 1120. The output of the combiner ~~1418~~ 1318 is provided to an input of the peak compressor 1302, and an output of the peak compressor 1302 is provided to the input of the bass punch unit 1120.

Please replace the paragraph beginning on page 33 line 15 with the following:

Although some embodiments are described herein with reference to various sound enhancement system, the invention is not so limited, and can be used in a variety of other contexts in which it is desirable to adapt different embodiments of the sound enhancement system to different situations. ~~To facilitate a complete understanding of the invention, the remainder of the detailed description is organized into the following sections and subsections:~~

Please replace the paragraph beginning on page 33 line 21 with the following:

Figure 15 is a block diagram 1500 of a differential perspective correction apparatus 1502 from a first input signal 1510 and a second input signal 1512. In one embodiment the first and second input signals 1510 and 1512 are stereo signals; however, the first and second input signals 1510 and 1512 need not be stereo signals and can include a wide range of audio signals. As explained in more detail below, the differential perspective correction apparatus 1502 modifies the audio sound

information, which is common to both the first and second input signals 1510 and 1512 in a different manner than the audio sound information, which is not common to both the first and second input signals 1510 and 1512.

Please replace the paragraph beginning on page 35 line 3 with the following:

Figure 16 is an amplitude-versus-frequency chart, which illustrates the common-mode gain at both the left and right output terminals 1530 and 1532. The common-mode gain is represented with a first common-mode gain curve 1600. As shown in the common-mode gain curve 1600, the frequencies below approximately 130 hertz (Hz) are de-emphasized more than the frequencies above approximately 130 Hz. ~~For frequencies above approximately 130 Hz, the frequencies are uniformly reduced by approximately 6 decibels.~~

Please replace the paragraph beginning on page 35 line 10 with the following:

Figure 17 illustrates the overall correction curve 1700 generated by the combination of the first and second cross-over networks ~~2106~~1520, and ~~2107~~1522. The approximate relative gain values of the various frequencies within the overall correction curve ~~1300~~ 1700 can be measured against a zero (0) dB reference.

Please replace the paragraph beginning on page 35 line 14 with the following:

With such a reference, the overall correction curve 1700 shows two turning points labeled as point A and point B. At point A, which in one embodiment is approximately ~~2125~~ 170 Hz, the slope of the correction curve changes from a positive value to a negative value. At point B, which in one embodiment is approximately ~~21.8~~2 kHz, the slope of the correction curve changes from a negative value to a positive value.

Please replace the paragraph beginning on page 35 line 20 with the following:

Thus, the frequencies below approximately ~~2125~~ 170 Hz are de-emphasized relative to the frequencies near ~~2125~~ 170 Hz. In particular, below ~~2125~~ 170 Hz, the gain of the overall correction curve 1700 decreases at a rate of approximately 6 dB

per octave. This de-emphasis of signal frequencies below ~~2125~~ 170 Hz prevents the over-emphasis of very low, (i.e. bass) frequencies. With many audio reproduction systems, over emphasizing audio signals in this low-frequency range relative to the higher frequencies can create an unpleasurable and unrealistic sound image having too much bass response. Furthermore, over emphasizing these frequencies may damage a variety of audio components including the loudspeakers.

Please replace the paragraph beginning on page 35 line 29 with the following:

Between point A and point B, the slope of one overall correction curve is negative. That is, the frequencies between approximately ~~2125~~ 170 Hz and approximately ~~21.8~~2 kHz are de-emphasized relative to the frequencies near ~~2125~~ 170 Hz. Thus, the gain associated with the frequencies between point A and point B decrease at variable rates towards the maximum-equalization point of -8 dB at approximately ~~21.8~~2 kHz.

Please replace the paragraph beginning on page 36 line 4 with the following:

Above ~~21.8~~2 kHz the gain increases, at variable rates, up to approximately ~~120~~ 20 kHz, i.e., approximately the highest frequency audible to the human ear. That is, the frequencies above approximately ~~21.8~~2 kHz are emphasized relative to the frequencies near ~~21.8~~2 kHz. Thus, the gain associated with the frequencies above point B increases at variable rates towards ~~120~~ 20 kHz.

Please replace the paragraph beginning on page 36 line 25 with the following:

Equalization of the differential signal in accordance with the overall correction curve 1700 de-emphasizes the signal components of statistically lower intensity relative to the higher-intensity signal components. The higher-intensity differential signal components of a typical audio signal are found in a mid-range of frequencies between approximately 2 kHz to 4 kHz. In this range of frequencies, the human ear has a heightened sensitivity. Accordingly, the enhanced left and right output signals produce a much improved audio effect.